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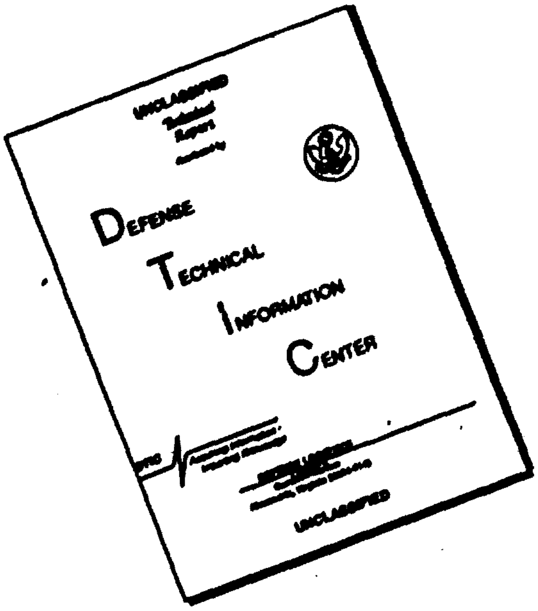

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A NEW METHOD OF TELETYPE MODULATION

DONALD J. GRAY

22 SEPTEMBER 1952

TECHNICAL REPORT NO. 9

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A NEW METHOD OF TELETYPE MODULATION

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ABSTRACT

In order to overcome the problems of frequency selective fading in radio teletype communication, a modulation method is proposed which distributes the transmitted energy over a band of frequencies instead of concentrating the energy around one or two discrete frequencies. Use is made of the matched filter method of reception. It is shown that signal-to-noise ratio at the receiver output is independent of the bandwidth occupied by the transmitted signal. The effective noise bandwidth of the system can be made equal to the minimum bandwidth required by the information transmitted without regard to the channel bandwidth occupied. A laboratory model of the system is described.

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CONTENTS

A NEW METHOD OF TELETYPE MODULATION*

INTRODUCTION

For the past fifteen months the possibilities of VHF propagation beyond the horizon have been under investigation at Lincoln Laboratory of M.I.T. and at the Central Radio Propagation Laboratory of the National Bureau of Standards.[1] It has been found possible to receive signals reliably at distances as great as 1,000 miles at frequencies as high as 50 Mc. The received signal strength is small, and high-gain antennas and high-power transmitters are required. The signal is subject to the usual fading associated with sky-wave propagation. In addition to fading, the signal suffers interference from meteor reflections. High-speed meteors traversing the transmission path leave trails of ionized gas which reflect the signal. Since the ionization centers of these trails move with the meteors, there is a Doppler frequency shift in the signal received by way of meteor reflection. The signal returned from meteor trails is usually much stronger than that propagated by the usual mechanism.

Attempts have been made to transmit teletype over these VHF links; frequency-shift keying has been the system of modulation used. Because of the fading of the signal it has been necessary to employ space diversity, using two receiving antennas and automatically selecting the output from the antenna that momentarily has the best signal. The result of the Doppler effect caused by passing meteors has been to shift the received signal up or, occasionally, down in frequency. If the Doppler shift happens to correspond to the teletype-shift frequency, the system will make errors because a pulse transmitted at the lower frequency will be shifted to the upper frequency.

It has been decided to investigate the possibilities of a modulation system for teletype which distributes the transmitted energy uniformly over a frequency band. It was felt that the operation of such a system would be less adversely affected by fading and by meteors than would frequency shift teletype; frequency shift modulation lumps the transmitted energy around two discrete frequencies and fading at either frequency causes errors. The work reported here is the result of a proposal by Professor R. M. Fano of M.I.T. and has been in progress since January of 1952.

A MATCHED FILTER SYSTEM OF TELETYPE MODULATION

The usual method of modulation used for transmitting teletype messages by radio is a binary form of frequency modulation known as frequency shift keying, or FSK. In FSK, one radio frequency is designated as the mark frequency and another as the space frequency. It is common practice to separate mark and space frequencies by about 800 cps. When the teletype machine is generating mark pulses the RF carrier frequency is shifted to the mark frequency. For space pulses the frequency is shifted to the space frequency. It will be shown that this system of modulation is quite immune to noise interference but is adversely affected

*This report is identical with a thesis of the same title submitted in partial fulfillment of the requirements for the Degree of Master of Science in the Department of Electrical Engineering at the Massachusetts Institute of Technology, May 16, 1952.

by fading in the transmission path.

Let us consider a teletype transmission consisting of alternate marks and spaces. To a very good approximation, the frequency spectrum of such a signal can be considered as that of two carriers separated by the shift frequency and one hundred per cent amplitude modulated by square waves 180° out of phase. The spectrum of such a signal is shown in Fig. 1 below. Since the duration of one teletype mark or space pulse is 22 milliseconds (msec) the energy in the signal is very closely bunched around the shift frequencies. The best linear filtering, assuming usual receiving techniques, is accomplished by a filter with the frequency characteristic shown in Fig. 2. It is usual to follow such a filter with a limiter-discriminator

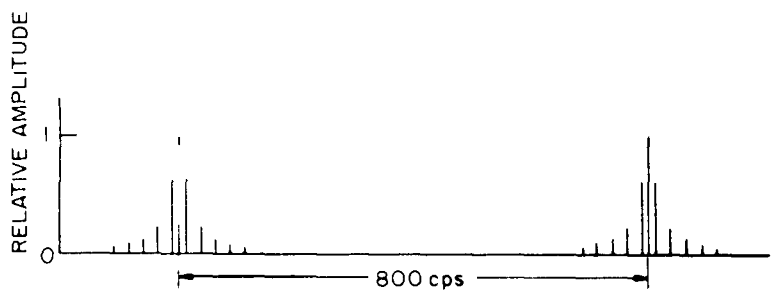


Fig. 1. Spectrum of frequency shift teletype transmitting alternate marks and spaces.

type detector. However, the noise bandwidth is determined by the filter and is of the order of 100 cps. In practice it is difficult to make a filter as narrow as this, and it is not usually desirable to do so. Changes in transmitter and receiver frequencies may necessitate a receiver filter considerably wider than that shown in Fig. 2.

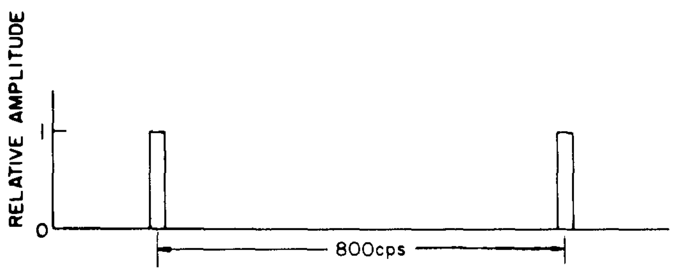


Fig. 2. Frequency characteristic of an ideal filter for FSK.

In any form of radio transmission propagated by mechanisms other than ground wave or direct ray, one experiences fading at the receiver. This fading, usually incoherent in nature, is caused by signals traveling from transmitter to receiver by many paths simultane-

ously. The relative attenuations and time delays of these paths vary slowly and in random fashion. Consequently, the signal at the receiving antenna, which is the vector sum of signals arriving by all paths, varies in amplitude also in a random fashion. It sometimes happens that the phases and amplitudes of the signals at the antenna give a vector sum of nearly zero. When this happens, the signal falls below the ambient noise and the communication system breaks down. It is possible for serious fades to occur as often as once every few seconds, even when the average signal is considerably above the noise.

Since fading of the type described is caused by several signals negating each other at the receiver, it is obvious that its occurrence is intimately related to both frequency and path length. In practice, the problem of fading has been met by a technique known as space-diversity reception. In space-diversity operation, one erects two or more receiving antennas separated by distances large with respect to the operating wavelength. Each antenna feeds a receiver, and a switching circuit takes output from the receiver having the best signal. Since there is little correlation of fading among the several antennas it is possible by space diversity to reduce the probability of error caused by fading to an arbitrarily low value.

The long-range VHF propagation studies currently under way at the M.I.T. Lincoln Laboratory have required very large receiving antennas. Since these antennas are expensive and space consuming, and since radio frequency spectrum in the VHF region is not at nearly the premium that it is in the HF region, it would be desirable to devise some scheme for teletype modulation which would avoid the need for space diversity by using what might be called frequency diversity. Since fading is caused by phase cancellation it is obvious that it can be severe only at discrete frequencies at any one time.

This theory has been verified by experimental evidence obtained from frequency-modulation tests. The 50 Mc. transmitter at Cedar Rapids, Iowa, was frequency modulated with a low audio frequency. The carrier was deviated ± 75 kc. The audio-frequency voltage from an FM receiver at South Dartmouth, Massachusetts, was observed on an oscilloscope. When fading occurred, noise appeared at only one or two values of instantaneous modulating voltage, indicating that the fading was indeed frequency selective in nature. An oscillogram showing this phenomenon appears in Fig. 3.[2]

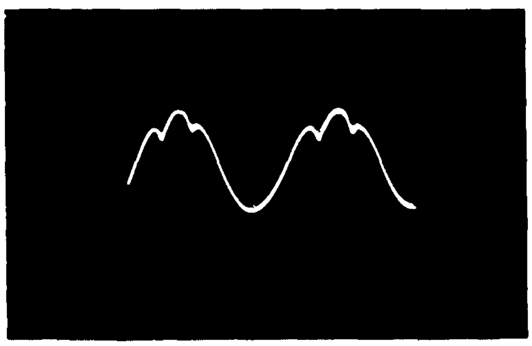


Fig. 3. Low frequency sinusoidal frequency modulation showing frequency selective fading.

Since fading does occur only at discrete frequencies, a system of teletype modulation which distributes the transmitted energy over a fairly wide frequency band instead of lumping it at two points, as does FSK, should be relatively immune to errors caused by fading.

A system of modulation proposed by Professor R. M. Fano, which is the subject of this report, does spread the transmitted energy over a frequency band without increasing the noise bandwidth of the receiver. According to the theorem of D. O. North, Van Vleck, and Middleton,[3,4] the optimum linear receiving filter for determining the presence or absence of a transmitted pulse is one whose system function, $H(\omega)$, is the conjugate of the Fourier transform of the time function of the transmitted pulse.

The system shown in Fig. 4 is an optimum system in that no other linear receiving

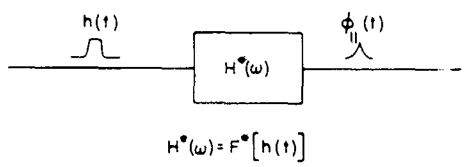


Fig. 4. An optimum receiving filter.

filter can improve the signal-to-noise ratio. The transmitted pulse, $h(t)$, can be considered to be the impulse response of a filter whose system function is $H^*(\omega)$, since $H(\omega)$ is the conjugate of the transform of $h(t)$. However, if $h(t)$ is the impulse response of a filter with system function $H^*(\omega)$, the impulse response of the filter with system function $H(\omega)$ is $h(-t)$. Hence,

$$H(\omega) = \int_{-\infty}^{\infty} h(t)\, e^{-j\omega t}\, dt \quad , \qquad (1)$$

$$H^*(\omega) = \int_{-\infty}^{\infty} h(t)\, e^{j\omega t}\, dt \quad . \qquad (2)$$

Let $t = -\tau$,

$$H^*(\omega) = \int_{\infty}^{-\infty} h(-\tau)\, e^{-j\omega\tau}\, (-d\tau) \quad ,$$

$$= \int_{-\infty}^{\infty} h(-\tau)\, e^{-j\omega\tau}\, d\tau \quad . \qquad (3)$$

This is of the same form as

$$H^*(\omega) = \int_{-\infty}^{\infty} h(-t)\, e^{-j\omega t}\, dt. \qquad (4)$$

A communication system capable of transmitting the on-off type of information can be made as follows. An arbitrary receiving filter is constructed. Its impulse response, $h(t)$, is determined and read into a storage unit. The storage unit is then taken to the transmitter. When it is desired to transmit a pulse, $h(t)$ is read out backwards from the storage unit and transmitted as $h(-t)$. The output of the receiving filter $S_o(t)$, is then the autocorrelation function of $h(t)$ in real time instead of in T.

$$S_o(t) = \int_{-\infty}^{\infty} S_i(t - \tau)\, h(\tau)\, d\tau \quad , \tag{5}$$

$$S_i(t - \tau) = h(\tau - t) \quad , \tag{6}$$

$$S_o(t) = \int_{-\infty}^{\infty} h(\tau - t)\, h(\tau)\, d\tau \quad . \tag{7}$$

As with any autocorrelation function, this function $\phi_{11}(t)$, has a maximum at $t = 0$ of value

$$\int_{-\infty}^{\infty} h^2(\tau)\, d\tau \quad . \tag{8}$$

This integral is proportional to the energy in the transmitted pulse. The rms noise at the output of the receiving filter, assuming additive white noise at the input of spectral power density K per radian per second, is

$$\overline{n_o^2} = \int_{-\infty}^{\infty} K\, |H(\omega)|^2\, dw \quad , \tag{9}$$

$$\begin{aligned} K \int_{-\infty}^{\infty} |H(\omega)|^2\, d\omega &= K \int_{-\infty}^{\infty} H(\omega)\, H^*(\omega)\, d\omega \quad , \\ &= K \int_{-\infty}^{\infty} d\omega\, H(\omega) \int_{-\infty}^{\infty} h(t)\, e^{j\omega t}\, dt \quad , \\ &= K \int_{-\infty}^{\infty} dt\, h(t) \left[\int_{-\infty}^{\infty} d\omega\, H(\omega)\, e^{j\omega t} \right] \quad , \qquad (10) \\ &= 2\pi K \int_{-\infty}^{\infty} h^2(t)\, dt \end{aligned}$$

$$\sqrt{\overline{n_o^2}} = \sqrt{2\pi K \int_{-\infty}^{\infty} h^2(t)\, dt} \quad . \tag{11}$$

Therefore, the ratio of peak signal to rms noise at the output is

$$\sqrt{\frac{\int_{-\infty}^{\infty} h^2(t)\, dt}{2\pi K}} \quad . \tag{12}$$

It is seen, therefore, that the voltage signal-to-noise ratio at the output of the receiving filter is proportional to the square root of the energy in the transmitted pulse and is

otherwise independent of pulse shape. The important conclusion to be drawn from this is that the filter can be designed to have an impulse response with the desired broad frequency spectrum without causing signal-to-noise ratio deterioration at the receiver.

We can now compare the performance of the proposed system with that of frequency-shift modulation in the presence of additive noise. Let us consider a signal from the antenna that cannot exceed A in instantaneous voltage and a noise which has the characteristics given by the equation $\overline{n_o^2} = Kw$, where w is the bandwidth in radians per second. The ratio of peak signal to rms noise for the correlation filter has been shown to be

$$\sqrt{\frac{\int_{-\infty}^{\infty} h^2(t)\,dt}{2\pi K}} \quad . \tag{12}$$

If D is the duration of h(t), the maximum possible value that this ratio can have is

$$\sqrt{\frac{A^2D}{2\pi K}} \quad . \tag{13}$$

If D is now the teletype period and if we consider the case of FSK using the standard shift frequency of 800 cps, we get the following: the mark channel, we can assume, has an AM square wave of period 2D, and amplitude A is put through an ideal rectangular filter of bandwidth $2/2D$. The peak signal voltage from this filter is $4A/\pi$. The rms output noise voltage is

$$\sqrt{\frac{2K\pi}{D}} \quad . \tag{14}$$

The output peak signal to rms noise ratio is then

$$\frac{4}{\pi}\sqrt{\frac{A^2D}{2\pi K}} \quad . \tag{15}$$

It is seen, therefore, that the proposed system is about on a par with FSK with respect to additive noise.

If the correlation system is to be used for teletype communication it is necessary to have a filter with an impulse response of about 20 msec duration. In order to transmit the maximum energy per pulse it is desirable that the envelope of h(t) be as nearly rectangular as possible. The frequency spectrum of h(t) should be as broad and as flat as possible while consistent with channel bandwidth limitations. For the VHF study a bandwidth of 50 kc seems about right.

Because of the foregoing requirements it was decided that, at least initially, an acoustic filter should be tried. Such a filter would be cheaper and more quickly built than an equivalent electrical filter. For purposes of experimentation it was decided to use a magnetic tape recorder for the storage unit. Such a storage unit provides an easy method of reversal of h(t) by physically turning the tape around and playing it in the direction opposite to that used in recording.

EXPERIMENTAL PROCEDURE AND RESULTS

The initial experiment in this study was performed by Professor Fano. A room was filled with reflecting sheets of metal, a loudspeaker placed in one corner and a microphone in another. The loudspeaker was excited with short pulses and the microphone output recorded. When the recording was played backwards into the loudspeaker the microphone output had the appearance of an auto-correlation function; it was an even function with a sharp maximum at the center.

When it was decided to pursue this subject further, the type of filter to be constructed was discussed. An acoustic delay line with a number of taps seemed to be a reasonable approach to the problem. The first trial model to be built, shown in Fig. 5, consisted of a short piece of S-band wave guide with a loudspeaker driver unit mounted on one end and the other end packed with absorbent cotton. Five small tubes leading to a microphone were connected to the side of the wave guide at random positions along its length. The impulse response of this filter and the auto-correlation function of the impulse response are shown in Fig. 6.

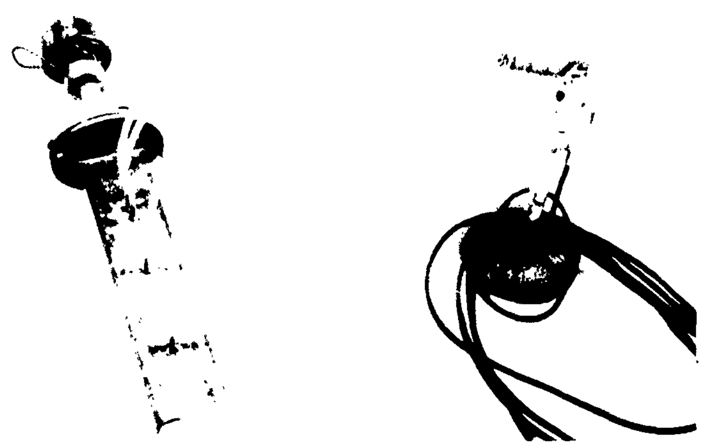
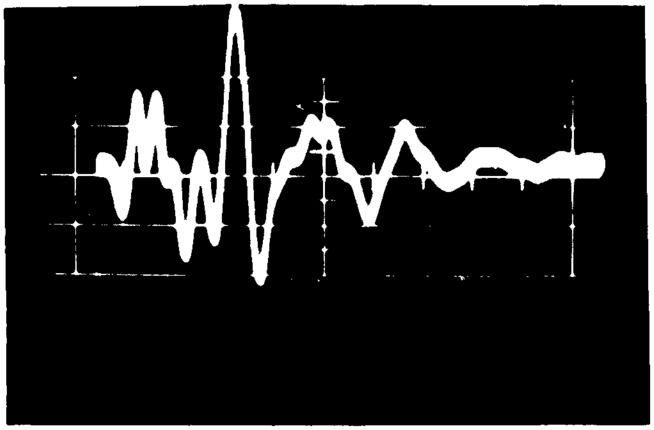


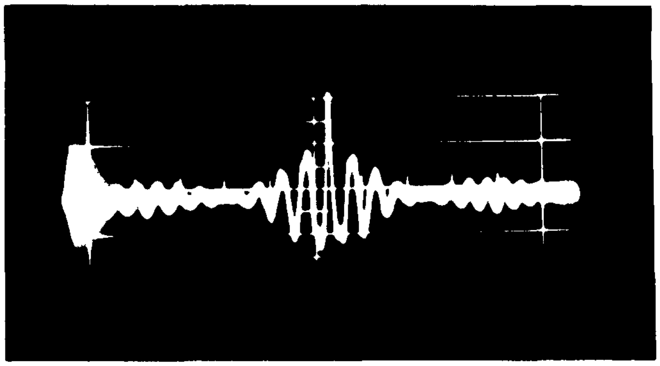
Fig. 5. Early filter constructed of S-band waveguide.

At this point it was decided to build a full scale model of the filter, one which would have an impulse response lasting for 20 msec and which would have as wide and as flat a spectrum as possible using audio components. The filter was built as shown in Fig. 7. The input transducer used was the driver unit from a commercial high-frequency tweeter. An exponential matching transformer was used to match the driver unit to the steel tubing.

The output transducer was a crystal microphone manufactured by Massa Laboratories.[5] The frequency response of this microphone is flat to 20 kc. The ends of the tubes opposite to the transducers were terminated with tapered plugs of Fiberglas board to prevent reflections. Coupling between the two tubes was accomplished by drilling holes in the mating surfaces.



(a) Impulse response h(t) of early filter



(b) Autocorrelation function $\phi_{11}(t)$ for early filter

Fig. 6. Impulse response and its autocorrelation function of the early wave-guide filter.

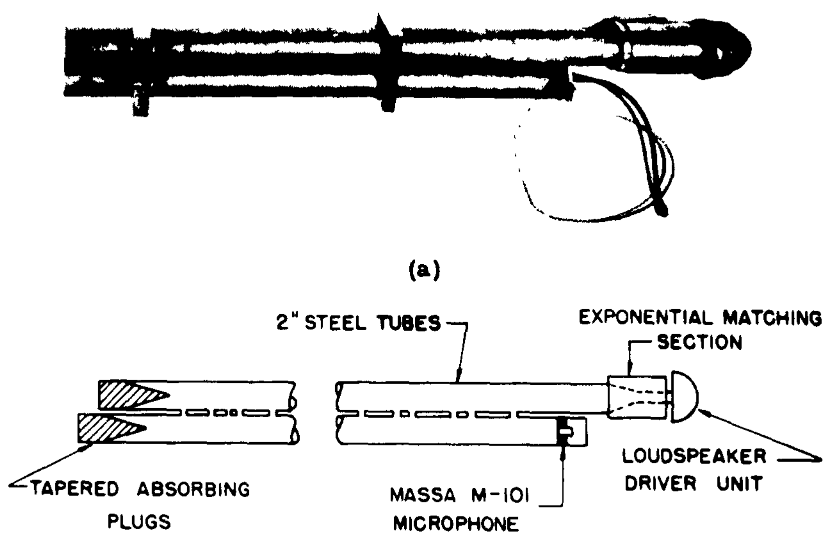


Fig. 7. Later model of filter constructed of two-inch steel tubing.

The character of h(t) is determined largely by the number and placement of the coupling holes. It was decided that a spectrum flat to 5 kc was all that could reasonably be expected of the impulse response of the filter because of the limitations of the audio devices used. If flatness much beyond 5 kc were to be achieved, it would have been necessary to use steel tubing of small size in which the attenuation would have been excessive. Since the desired impulse response was to last for 20 msec and was to have frequency components to 5 kc, it was decided to use 100 coupling holes. In the matter of placement of these holes, a logical method seemed to be a random spacing. To this end 100 numbers out of a possible 10,000 were selected from a table of random numbers.

It was desired to check the spectrum of h(t) before the holes were drilled. Rather than compute the spectrum from the hole placement – a laborious task – it was decided to construct an idealized h(t) and measure its spectrum. For this purpose a magnetic drum storage unit available in the laboratory was used. An indexing card with the hole spacing arranged around the circumference of a circle was made and fastened to the drum. The drum was then turned by hand and a pulse recorded at every position corresponding to a hole.

After one hundred pulses corresponding to the one hundred coupling holes had been recorded, the drum was run at 3600 rpm. Following suitable slicing to remove noise, the spectrum of the idealized h(t) was measured. The duration of the pulses comprising h(t) after the slicer was about 10 μsec and the pulses can therefore be considered impulses at frequencies below 10 kc. The integrated spectrum of the idealized h(t) is presented in Fig. 8. Each point on the curve is the average of five adjacent measured points. This averaging was done in order to make the curve easier to interpret. The curve is scaled in frequency since

the period of rotation of the drum was 16.7 msec and the desired duration of h(t) was 20 msec. The flatness of spectrum shown by this curve indicated that the hole placement was satisfactory. The hole placement used is contained in Appendix I.

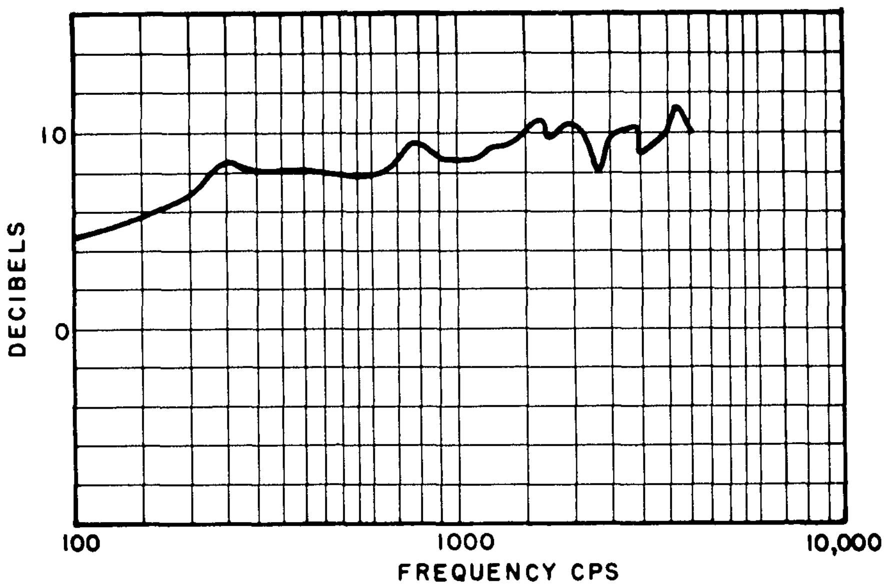


Fig. 8. Spectrum of idealized h(t) of later filter.

Because the Massa microphone has low sensitivity it was desired to present as high a sound level as possible to the microphone. Therefore it would seem desirable to make the coupling holes large. In the interest of ease in machining, it was decided that all the coupling holes should be of the same size. Since each coupling hole extracts a small amount of energy from a pulse traveling past it in either tube, the size of the holes is limited if the power in h(t) is to be uniform throughout its duration. The holes were made large enough so that a pulse traveling from driver unit to microphone by way of the farthest hole would be attenuated by 3 db with respect to a pulse traveling by way of the nearest hole.

Effect of Noise

The impulse response of the filter, h(t), is shown in Fig. 9. The duration of the exciting pulse was reduced to the point where further reduction caused no significant change in the appearance of h(t). The pulse duration was then about 50 μsec. The duration of h(t) in this oscillogram is 20 msec. A General Radio Company Wave Analyzer, type 736A, was used to measure the spectrum of h(t). The integrated spectrum of h(t) is presented in Fig. 10. Each point on the curve is the average of nine adjacent measured points.

The impulse response h(t) was recorded using an Ampex 3066-B magnetic tape recorder which has a frequency response flat within 3 db from 300 cps to 70,000 cps and so is entirely adequate for this use. The play-back of h(–t) was used to excite the driver unit.

The autocorrelation function $\phi_{11}(t)$ appearing at the output of the filter is shown in Fig. 11.

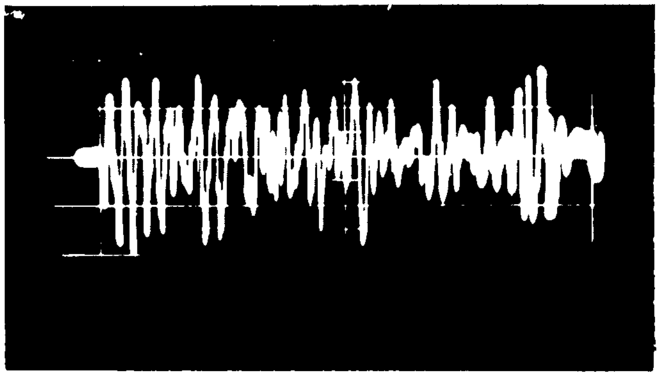


Fig. 9. Impulse response h(t) for filter shown in Fig. 7.

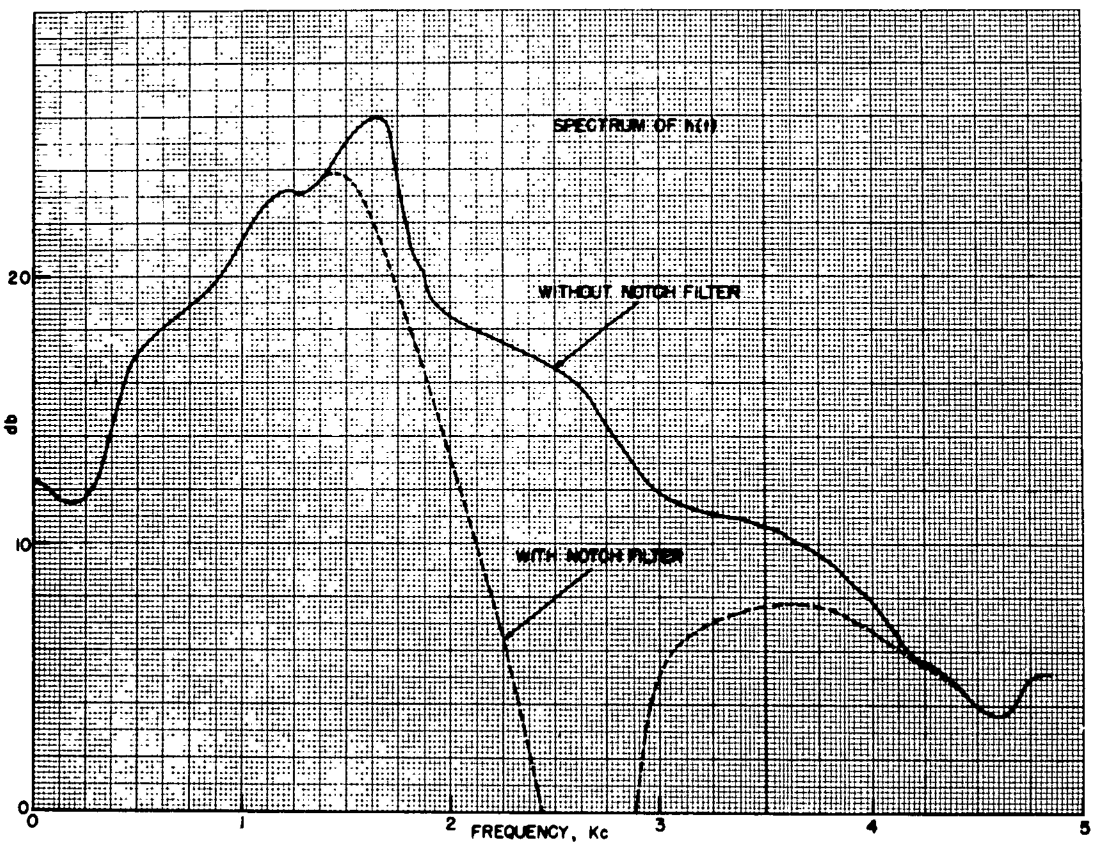


Fig. 10. Spectrum of h(t) shown in Fig. 9.

Fig. 11. $\phi_{11}(t)$ of Fig. 9.

In order to predict the operation of a modulation system of this type over the VHF communication links under study, it was decided to perform experiments in the laboratory which would simulate actual operation. One rather obvious experiment was the determination of the behavior of the system in the presence of additive noise. The noise generator used was of the gas diode type. A 40 db per decade, low-pass RC filter with its break point at 7.5 kc was used to limit the output of the noise generator to the frequency range of interest. The spectrum of the noise is shown in Fig. 12. A block diagram showing the experimental set-up is shown in Fig. 13. A thermal milliameter, driven by an audio power amplifier, was used to measure the input signal-to-noise ratio. In all cases the duty ratio of h(−t) was 50 per cent. It was necessary to use this duty ratio because the duration of $\phi_{11}(t)$ is twice that of h(t), and a duty ratio of greater than 50 per cent would have caused interference effects at the extremes of $\phi_{11}(t)$. In all experiments the input noise was increased until a Tektronix oscilloscope could no longer synchronize reliably on the maximum of $\phi_{11}(t)$. It was found that the maximum input noise that could be tolerated before the oscilloscope lost synchronization was 12 db above the input signal. An oscillogram of the output from the acoustic filter with this amount of noise at the input is shown in Fig. 14.

Slot Rejection Filter

In order to simulate the effect of frequency selective fading in the transmission path, it was decided to use a narrow rejection, or notch filter, in the signal channel. The circuit diagram of this filter is shown in Fig. 15, and its frequency response is shown in Fig. 16. The experimental setup was as shown in the block diagram of Fig. 17. With the notch filter in the signal channel, using the same criterion of maximum noise as above, it was found that the input signal-to-noise ratio was −7.5 db. Oscillograms are shown in Fig. 18.

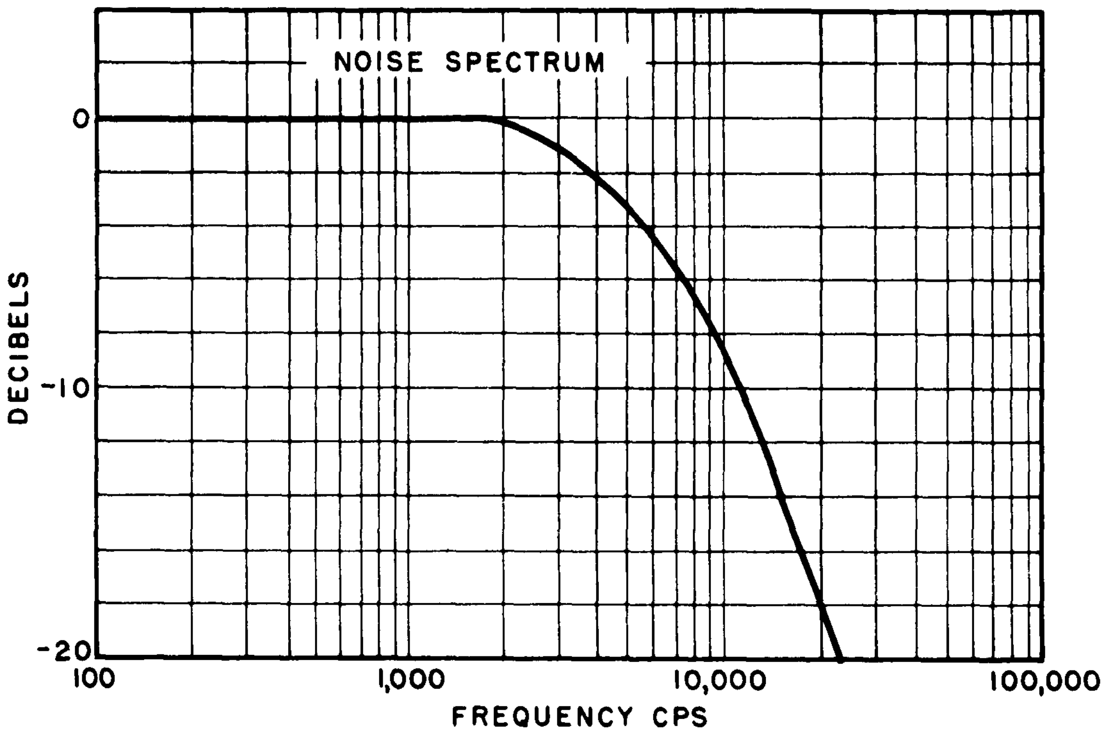


Fig. 12. Spectrum of noise used in experiments.

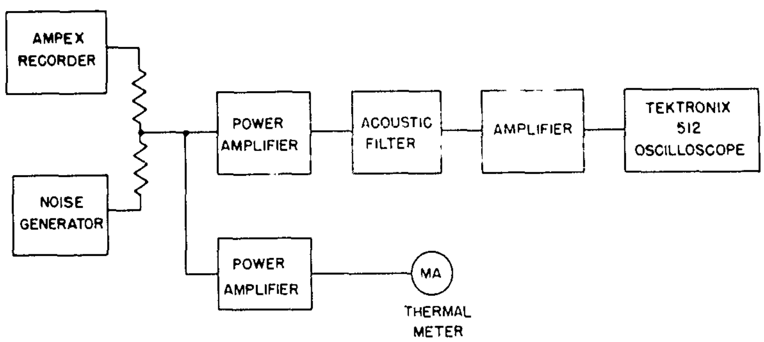


Fig. 13. Experimental setup.

Fig. 14. $\phi_{11}(t)$ with input noise 12 db above signal.

Limiting of h(−t)

If h(t) were to be transmitted by radio, one method of modulation would be to shift the spectrum of h(t) upward in frequency and use it to excite the transmitter. Transmitters built with cascaded class C amplifiers are cheaper and more readily obtainable than are linear amplifiers, particularly if the power requirement is high. If h(t) were used to excite a class C transmitter, however, the amplitude variations would be lost. In order to determine the effect of such amplitude limiting, an experiment was performed in which h(−t) was amplitude limited before being applied to the acoustic filter. The circuit diagram of the limiter used is shown in Fig. 19. This limiter removed essentially all the amplitude variation of h(−t). The block diagram of Fig. 20 shows the experimental setup. It was found that the maximum allowable input noise was 10.5 db above the input signal measured after limiting. Oscillograms are shown in Fig. 21.

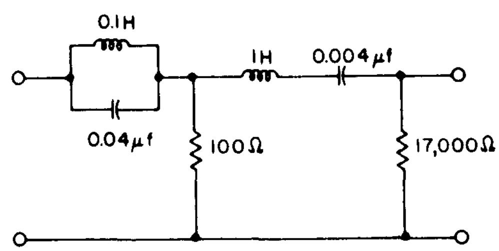


Fig. 15. Circuit of the notch filter.

Change of Time Scale

It was desired to determine the effect of variation in the velocity of sound in the acoustic filter. Rather than attempt to vary the velocity in the tubes, it was decided to vary the speed of the tape recorder. The recorder capstan motor was supplied with variable-frequency power. A speed variation of 0.6% was found to reduce the amplitude of $\phi_{11}(t)$ to 80 % of its maximum value. Speed variation of 0.9% reduced the amplitude to 50% of maximum. At a speed variation of 1.8 % the peak of $\phi_{11}(t)$ had disappeared. Oscillograms showing $\phi_{11}(t)$ for speed variations of 5% and 10% are shown in Fig. 22.

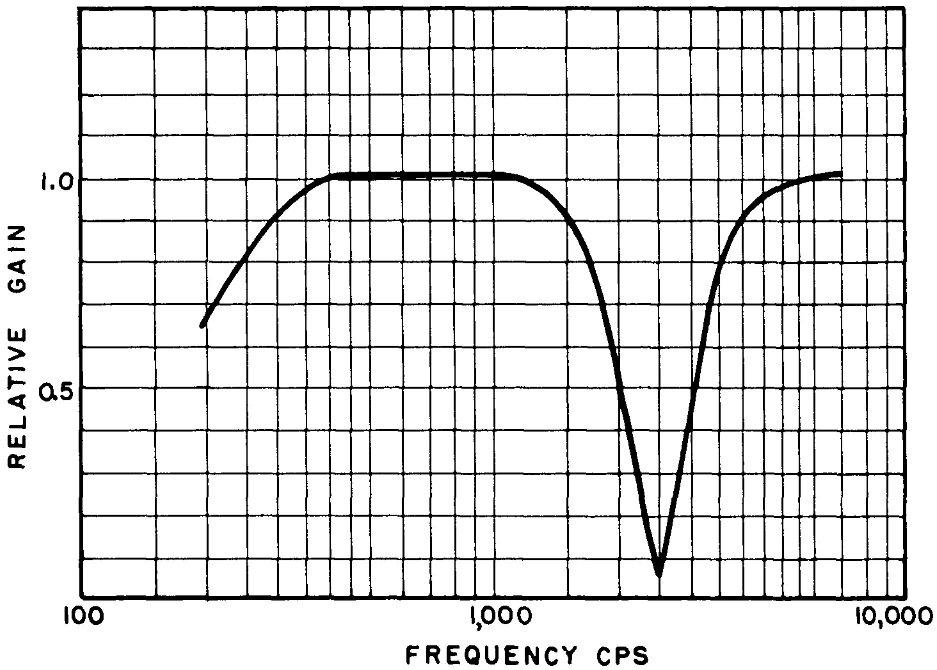


Fig. 16. Frequency response of the notch filter.

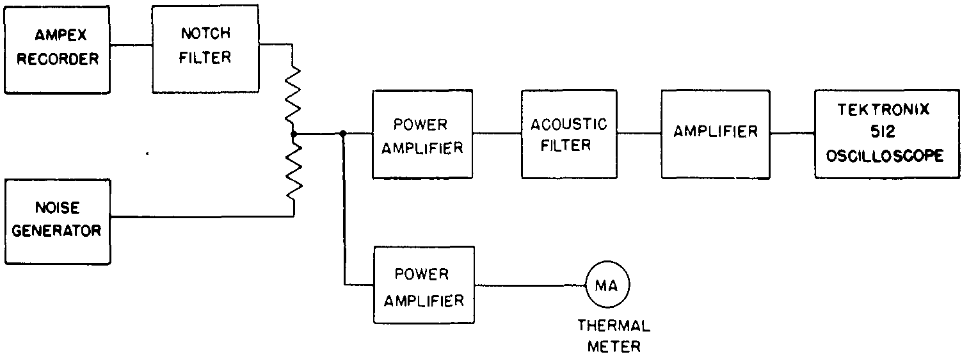


Fig. 17. Experimental setup with a notch filter in the signal channel.

(a) h(–t) after notch filter

(b) $\phi_{11}(t)$ after notch filter without noise

(c) $\phi_{11}(t)$ after notch filter with input noise 7.5 db above signal

Fig. 18. Oscillograms of experimental setup with a notch filter in the signal channel.

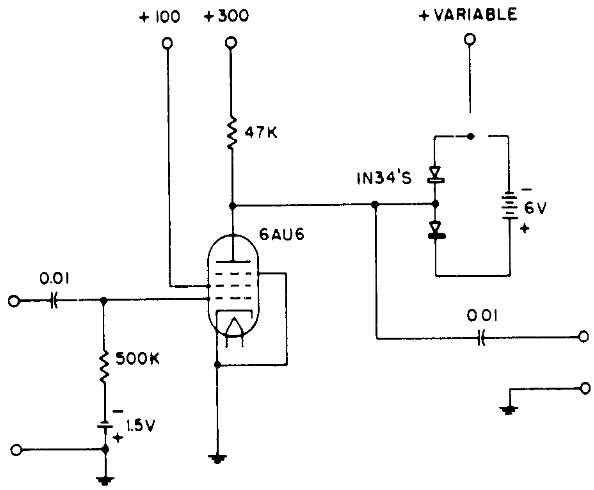


Fig. 19. Circuit of the limiter.

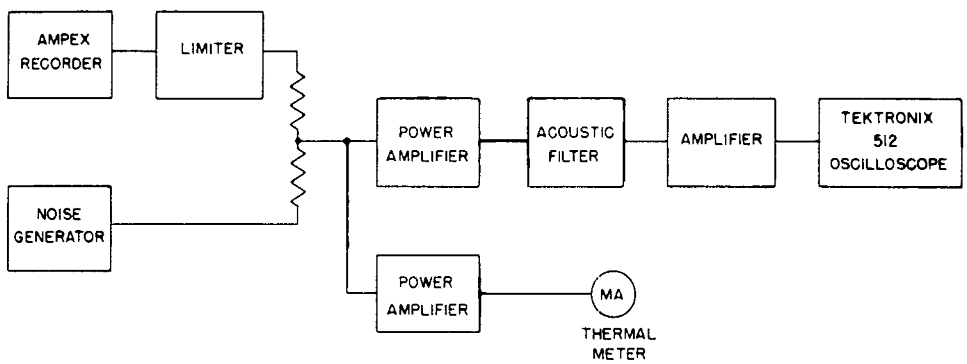
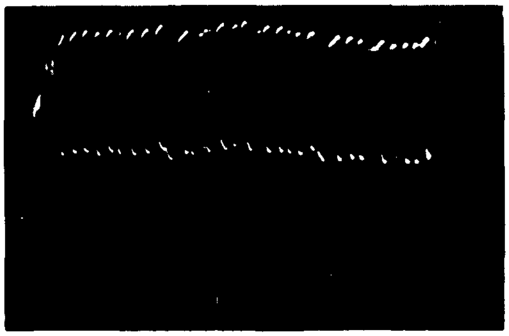


Fig. 20. Experimental setup with the limiter in the signal channel.



(a)

h(−t) amplitude limited

(b)

$\phi_{11}(t)$ after limiting of h(−t) without noise

(c)

$\phi_{11}(t)$ after limiting of h(−t) with input noise 10.5 db above signal

Fig. 21. Oscillograms of the experimental setup after the limiter in the signal channel.

(a)

$\phi_{11}(t)$, recorder running 5% fast

(b)

$\phi_{11}(t)$, recorder running 10% fast

Fig. 22. Oscillograms resulting from tape-recorder speed variations.

DISCUSSION OF EXPERIMENTAL RESULTS

Spectrum of Impulse Response

When the spectrum of h(t) of the filter is compared with that of the idealized h(t) described previously, it is apparent that a serious discrepancy exists. The spectrum of the impulse response of the filter is not nearly as flat as it was expected to be. It is thought that this non-flatness is caused by the loudspeaker driver unit and acoustic matching section. An obvious step that should be taken to improve the performance of the present filter is the installation of an equalizer to flatten the spectrum of h(t), at least to about 5 kc. This approach avoids the necessity of wondering what to do to the loudspeaker driver unit to improve its frequency response.

Rejection Filter

The experiment with the notch filter in the signal channel indicates that, as was expected, the system is not seriously affected by rejection of a small percentage of the signal spectrum. Because of this result, it would seem that this system of modulation would have definite advantages over FSK in the presence of multipath frequency-selective fading. The obvious disadvantage of the system – the fact that it requires large transmission bandwidth to transmit the small information rate of a teletype signal – is not at present too serious in the 30 to 50 Mc region, although it may become serious in the future.

Limiting of h(–t)

The experiment with limiting of h(–t) seems to indicate that much more information is contained in the phase function than in the amplitude function. It is noteworthy that removing all amplitude variation from h(–t) decreased the output signal-to-noise ratio by less than 2 db. The reason for this becomes apparent upon inspection of the expression for the maximum of $\phi_{11}(t)$. The integral

$$\int_{-\infty}^{\infty} h^2(t)\, dt \tag{16}$$

may be written

$$\int_{-\infty}^{\infty} h(t)h(t)dt \quad . \tag{17}$$

If h(–t), after limiting, is written h'(–t), the integral then becomes

$$\int_{-\infty}^{\infty} h(t)h'(t)dt. \tag{18}$$

Since $h'(-t)$ has only two possible values of instantaneous voltage and is positive when h(–t) is positive and negative when h(–t) is negative, one would not expect the value of the integral for the limited h(–t) to be greatly different from that for h(–t) not limited. It is fortunate that this is so, for it eases the transmitter design problem considerably. It is difficult and expensive to build a high-power, amplitude-modulated transmitter having modulation bandwidth capabilities greater than a few kilocycles. On the other hand, transmitters which would be suitable for use with the amplitude limited h(–t) are readily available.

Time Scale Stability Requirements

It is evident from the results of the experiment with varying recorder speeds that the characteristics of the filter used to generate h(t) must remain essentially constant if the system is to function properly. The storage unit used must also read out h(–t) with very little change in time scale from that at which h(t) was read in. In the case of the acoustic model described here, it is apparent that the greatest total change in time scale that can be tolerated is less than one per cent. If a filter is designed with a bandwidth of 50 kc the maximum allowable shift in time scale will be of the order of one-tenth of one per cent. It is certain, from these considerations, that the filter, however it is realized, must be rigidly constructed. Its temperature and possibly its humidity, if it is air-filled, must be closely controlled. The storage unit, if it is a magnetic drum, must have very good speed regulation, and the motor driving the drum must undoubtedly have a source of power with better frequency regulation than that of the power line.

COMMENTS AND RECOMMENDATIONS

Detection Problems

An improvement in signal-to-noise ratio would be obtained if it were known at the receiver when it was possible for a maximum of $\phi_{11}(t)$ to occur. If this information were available at the receiver, the output of the filter could be gated in such a way that the receiver only looked for a signal at times when a signal could be expected to be present. In this way the chances of a noise peak causing a false indication of a signal would be reduced. One such scheme would be a gating circuit synchronized to the pulse rate in such a way that if a maximum of $\phi_{11}(t)$ were present it would occur in the center of the gate pulse. The gate could then be made very narrow with a consequent improvement in system reliability.

The very fact that this system will operate with an input signal-to-noise ratio of less than unity rules out the use of any detector which discriminates against weak signals in favor of strong ones. It is therefore impossible to use any form of envelope detector such as a diode detector. Evidently synchronous detection must be used that will require very close control of the frequency of the local beating signal. A shift in the frequency of the beating signal will produce a corresponding shift in all the frequency components of h(–t).

Meteors

The effect of meteors will be to change not only the frequency but also the duration of h(–t). The greatest change in the duration of h(–t) to be expected is of the order of 0.004%. However, the spectrum of h(–t) can be shifted as much as 2000 cps by fast meteors if the system is operating at 50 Mc. The greatest meteoric frequency shifts observed during VHF propagation studies at Lincoln Laboratory have been of the order of 0.004%. The meteor problem will require more study before any definite conclusions can be reached about the effect of meteors on this system. It is possible that the filter will treat the signal returned from a meteor trail simply as noise and the system will continue to function on the signal propagated by the normal means.

Methods to Obtain More Bandwidth

It seems that 5000 cps is about all the bandwidth that can be expected from an air-filled acoustic filter. Above this frequency available audio components do not function as well as might be desired. Small tubing is required for the delay lines to prevent high-order modes of propagation. The attenuation of sound in air is proportional to frequency. Attenuation is increased as tube size is decreased. Consequently some other type of filter must be used if bandwidth of the order of 50 kc is required.

There are several possible approaches to the problem which show promise. A mercury delay line, constructed in similar fashion to the air-filled line used here, would be about 45 feet long for 20 msec duration of h(t). The bandwidth realized in the mercury line would be much greater than that of the air line. It is possible to obtain a good impedance match between quartz crystal transducers and mercury. Consequently the use of much higher frequencies is possible in mercury than in air. The attenuation of ultrasonic waves in mercury delay lines at a frequency of 10 Mc is about 5 db per millisecond.[6] Attenuation is proportional to the square of frequency. Therefore such a line operated with a 1 Mc carrier frequency would have attenuation sufficiently small to be negligible for this purpose. The fact that the line would be nearly 50 feet long is not necessarily too serious since it could be broken into several pieces with crystal transducers coupling the pieces together.

Another possibility is the quartz or crystalline metal line. It has been found that a pulse introduced into such a line is reflected internally by the crystal structure. The impulse response is of quite long duration compared to the delay of the line and appears to be very like noise in character. The possibilities of such devices are currently being investigated.

The author wishes to express his appreciation to Professor W. H. Radford for supervising the research reported here and to members of the staff of the Lincoln Laboratory for their valuable assistance and suggestions.

APPENDIX A

HOLE PLACEMENT

The following list of numbers specifies the spacing of 100 holes in the steel tubes. The distances are in inches measured from coincident references on each pipe. The holes are 0.11 in diameter.

1.572	42.096	76.884
1.968	44.352	77.640
6.156	46.560	80.928
6.280	46.764	81.060
6.648	48.432	82.200
10.008	48.560	83.436
10.440	49.848	84.540
10.560	50.244	87.972
12.036	50.370	88.560
14.088	51.444	90.576
14.268	51.586	90.700
15.096	51.720	91.488
15.456	52.032	94.056
16.116	52.920	94.560
17.844	53.892	97.296
18.828	57.060	97.728
19.236	59.448	99.552
20.340	60.060	100.464
21.864	60.384	104.340
22.044	62.328	104.712
22.788	64.488	105.852
26.532	65.256	108.048
27.636	66.600	108.732
27.780	66.900	111.600
30.852	67.476	112.560
31.404	70.236	113.100
32.160	70.812	113.230
32.316	70.930	113.784
32.976	71.172	114.324
39.096	71.916	115.464
39.348	74.124	116.232
40.044	74.448	116.496
40.716	74.676	116.772
		119.316

APPENDIX B

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